Digital Audio Broadcasting in the Short Wave Bands

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Abstract - For many decades the Short Wave broadcasting service has used high power, double-sideband AM signals to reach audiences far and wide. While audio quality was usually not very high, inexpensive receivers could be used to tune into broadcasts from distant countries. There was no incentive to switch to digital techniques because of several factors: the state-of-the-art in audio compression had not advanced sufficiently to provide good audio quality in the limited (10 kHz) bandwidth available; the propagation characteristics of the short wave propagation channel were known to be severe and hostile to digital signal transmission; and it was believed that digital receivers could not be manufactured at sufficiently low cost.

Lately, however, competition has started to build from other, often higher quality digital audio services such as satellite broadcasting and the Internet. These services are showing the superior quality that can be achieved by digital audio. Significant improvements in audio compression technology have been made and the cost of the digital processing circuitry required by digital audio receivers has come down, For these reasons, many of the world's major short wave broadcasters are now seriously investigating a conversion to digital audio broadcasting.

Propagation in the short wave (3-30 MHz) bands is the one factor that has not improved over time. The design of a robust digital audio broadcasting system involves many system trade-offs in the design of the signal structure, as well as in the receiver processing requirements. Limited bandwidth, a difficult propagation channel, and the desire to maximize the data rate all complicate the signal structure and the receiver. This paper describes the progress made to date on the development and testing of a system for a short wave digital audio broadcasting being developed at the Jet Propulsion Laboratory (JPL), under sponsorship of the Voice of America (VOA).

I. INTRODUCTION

'The short wave or HF (3 MHz to 30 MHz) bands. are suitable for long distance international broadcasting because radio signals at these frequencies are reflected from the ionosphere as well as from the surface of the Earth. This allows signals to travel thousands of kilometers by means of one, two, or more hops. The VOA, among other international broadcasters, is interested in the potential of digital audio in improving the quality and reliability of their broadcasting services. To take advantage of the existing transmitting facility infrastructure, it is hoped that this can be done using the existing high power HF transmitter assets. Better yet, the possibility of lower power requirements for digital broadcasting may lead to lower operating and eventual replacement costs for these transmitters.

The system for digital HF being developed at the Jet Propulsion is being adapted as far as possible from a design [hat was originally developed for satellite digital audio broadcasting, also under sponsorship of the VOA. Since the short wave propagation impairments are so different, some of the system trade-offs have to be redone. Some of the link impairment mitigation techniques that were developed for the satellite system, however, are turning out to be useful in a HF broadcasting system design.

This JPL project is onc of several studies currently under way to develop systems for HF digital broadcasting. It is hoped to select the best approach and define a "world standard" that could earn an endorsement from the ITU and would satisfy the requirements of the majority of short wave broadcasters.

11. SYSTEM DESIGN OVERVIEW

Short wave propagation over long distances works through signal reflection from the ionosphere, as well as from the ground in the case of multiple hops. Unfortunately the ionosphere is not an ideal reflecting surface. Its reflective properties vary with time of day, solar cycle, and other seemingly random phenomena, all of which results in a variety of propagation impairments. Listening to a DSB AM Short wave broadcast can range from the annoying to almost impossible because of these impairments, as well as from interference from other stations on the same frequency. Needless to say, a digital broadcasting system has to be robust in the presence of interference and able to overcome a variety of propagation problems.

The approach chosen for the VOA/JPL system is single carrier with coherent phase modulation. A moderate amount of error correction coding, together with equalization at the receiver, are the main techniques for combating the propagation impairments. The main thrust of the design effort is to maximize the data rate while maintaining an acceptable data error rate, thus acceptable audio. Factors such as modulation level, coding and equalizer training symbol overhead, and audio compression efficiency all enter into the trade-offs needed to optimize the system.

The 10 kHz bandwidth restriction limits the channel symbol transmission rate to about 8000 symbols per second, assuming suitable pulse shaping of the symbols. The audio bandwidth and quality requircn~ents/desires of HF broadcasters will require compressed audio data rates in the range of 16 kbps to 48 kbps. To transmit this over a 8 ksps channel will thus require a modulation scheme capable of 2 to 6 bits per symbol; exclusive of any synchronization or coding overhead.

The upper value of data rate may only be possible under very benign propagation conditions. Satisfactory operation at 16 kbps have been achieved with the experimental VOA/JPL system under moderately difficult propagation conditions, at a variety of transmitter power levels. Interestingly enough, increasing transmitter power above a certain point does not help under some propagation conditions.

The audio quality that can be achieved at a given data rate is dependent on the efficiency of the audio compression system. Important factors in the system design are the sensitivity of the audio decoder to bit errors and to their distribution. These factors affect the choice of the error correction code and its parameters. The interaction between the error correction capabilities of the receiver, the audio decoder, and the distortion tolerance of the listener can lead to a complex set of trade-offs. This was covered in some depth in a recent paper [1] and will not be discussed here further.

The audio compression system that has been tested with the VOA/JPL system is the AT&T G728 Low Delay CELP, which converts a 4 kHz audio spectrum into a 16 kbps data stream. It has good bit error tolerance, with error rates below 1OE-3 not being noticeable. As will be seen later, this results in a fairly low coding overhead requirement.

111. EXPERIMENTAL SYSTEM IMPLEMENTATION

The VOA/JPL system was initially implemented in software, simulating an 8000 symbol per second, 48 kHz sampled system, at an IF frequency of approximately 13 kHz. The three main modules consist of a modulator, a propagation channel simulator, and a receiver.

'f'he propagation channel simulator was used to optimize and test the receiver structure. Hardware interfaces were then developed to interface the modulator to the transmitters at the VOA transmitting facility in Delano, California. A short wave receiver was modified to allow signal reception and recording at remote locations. Most of the processing of the received signals is still done in non-real time on a PC.

The following is a description of the system components and the process of the initial definition of the signal structure.

A. Modulator

The modulator is capable of MPSK and MQAM modulation from BPSK to high orders of M. Included are options on error correction coding, insertion of training symbols for equalization, time interleaving, and setting up the transmission frame structure. The modulator accepts data from files which normally contain compressed audio data. The modulator is also capable of generating test signals for measuring propagation.

B. Receiver

A block diagram of the receiver structure is shown in Figure 1. The receiver is currently implemented in software, operating on a sampled input signal at an IF of 13 kHz. Coherent demodulation is accomplished on PSK and QAM modulated signals. A predictive decision-feedback (PDF), fractionally spaced equalizer (FSE) is used in the presence of multipath.

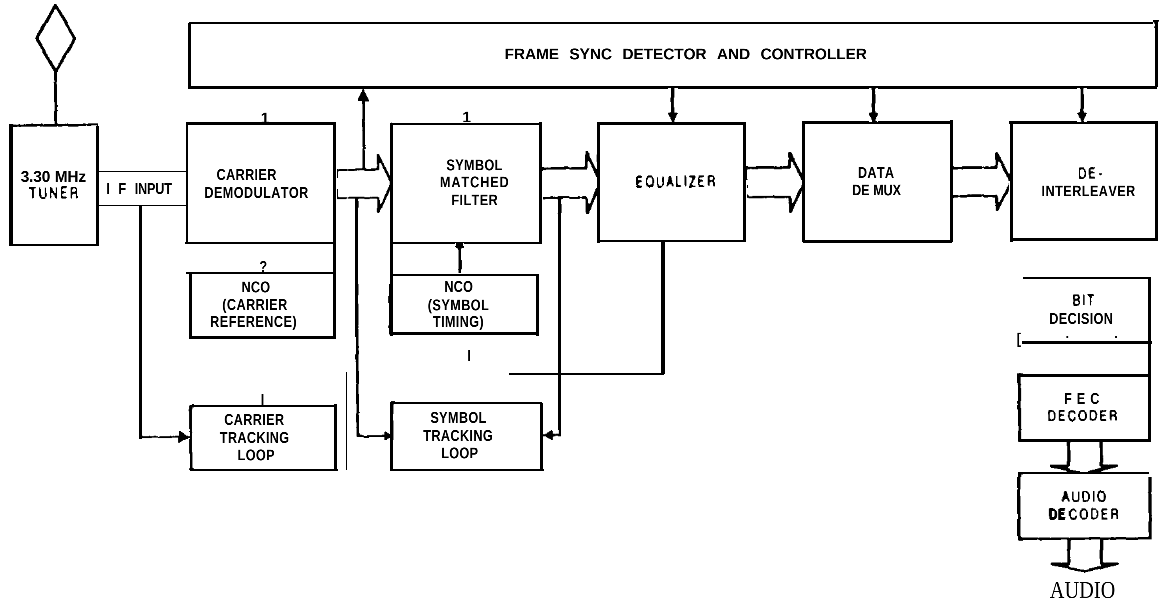


Figure 1. Receiver Block Diagram

C. Signal Structure

The signal structure consists of frame synchronization and data blocks. Frame sync is a 63 bit PN sequence with an extra bit added at the end to make it an integral number of bytes (8). The frame sync sequence is always modulated using BPSK. A data block can be an arbitrary number of bytes long, but should contain an integral number of audio frames. When Reed-Solomon coding is used, the data block length is set to have an integral number of audio frames and code blocks. When using a RS (255, 223) code the data block length is therefore 255*8*40 symbols or 255* M*8*40 data bits long. M is the number of bits per symbol in MPSK (3 for 8PSK) and 40 is the audio frame length in bytes.

Interleaving is done at the symbol level. When interleaving was used in the testing, the frame length was equivalent to the data block length (320 by 255 symbols). This results in a rather long interleaving delay (approx. 10 seconds) and would probably be made lower in a practical system.

A symbol from a known sequence of training symbols is inserted every N symbols into each data block. The value of N that was used for the field trials was 4 (training symbol ratio I out of 5 symbols).

D. Propagation Channel Simulator.

The HF propagation channel model is based on the tapped delay line approach. This allows an arbitrary number of signal components (representing multipath) to be individually modified by means of complex gain functions, according to the amplitude and phase transfer functions of each signal path, and combined to form the resultant signal.

The problem with this model is, of course, defining these transfer functions. Even from the limited number of propagation measurements performed during this task, it is obvious that the propagation characteristics of a given path can vary widely with time. It is useful to first optimize and test the system independently with the three classes of impairments which seem to be the major contributors to HF propagation difficulties. The effects that occur at the receiver can be divided into: signal power fluctuations, arrival of several signal multipath components, and interference from other signals.

E. Initial System Trades

Multipath is the first problem [o bc solved in a digital short wave receiver, which in our case is done through equalization. The propagation channel model is used to evaluate differences in the equalizer design parameters under some fixed, worst case conditions. An example of a widely used test condition is the three Rayleigh signal component case.

Once the equalizer is optimized, the channel simulator is used to trade-off other parameters, such as coding overhead requirements and achievable modulation complexity, as a function of the expected signal to noise ratio. This is first done under Gaussian channel conditions, as illustrated in Figure 2. This figure shows the relative power needed to achieve a given bit error rate with different modulation formats, without coding. Note that a doubling of the parameter M is needed to achieve a data rate increase of 8 kbps. The required transmitter power grows rapidly with modulation complexity. It is therefore very important to concentrate on lowering the data rate of the audio compression system.

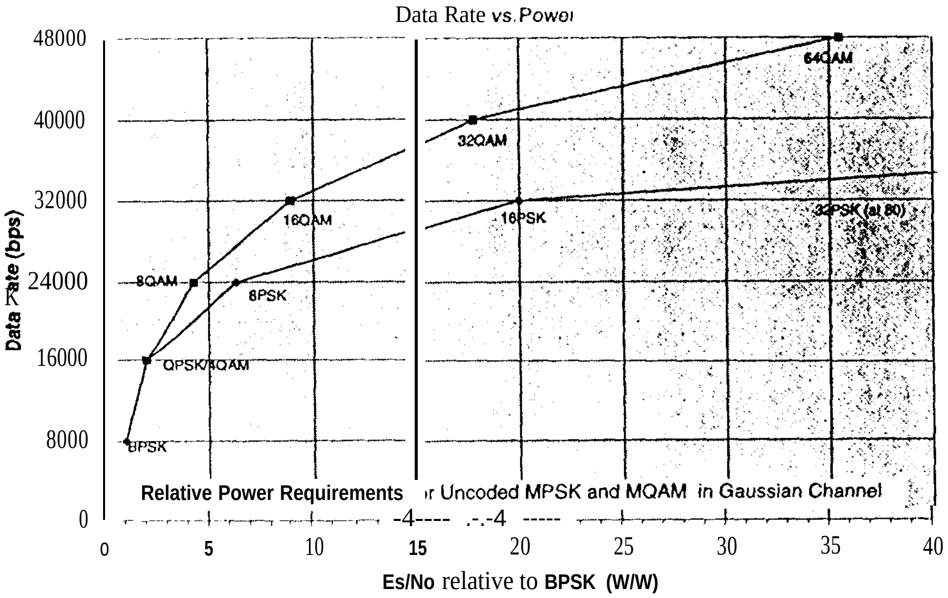


Figure 2. Required Transmitter Power vs. Achievable Data Rate at 8ksps Channel Rate.

The use of forward error correction (FEC) coding in a bandwidth limited channel has to be carefully considered. This is because the coding overhead subtracts directly from the information data rate. For example, a rate ½ **convolutional** code would reduce the data rate by a factor of 2. From Figure 2 it is evident that to restore the same information rate, it would be required to increase the modulation complexity by a large number. (E.g. 8PSK uncoded is equivalent to 64PSK coded, which would drive UP receiver complexity and still produce a negative performance gain).

There is however a need for some error correction coding because the fading channel produces an irreducible bit error rate floor, which can bc higher than the audio decoder can tolerate. The minimum amount of error correction that is needed is thus whatever it takes to reduce the bit error rate from the error floor to the error rate that the audio decoder can handle.

Figure 3 shows receiver performance under the three Rayleigh signal conditions, with 0.1 Hz Max Doppler. It can be seen that for modulation formats higher than QPSK, there is an error floor which cannot be lowered through increased power. For example for 8PSK, this error floor is at approximately 8E-3 bit error rate, which would be noticeable at the output of the AT&T G728 audio decoder. This error rate must therefore be lowered through the use of error correction coding to 1E-3 or lower, at which point audio impairments are no longer noticeable.

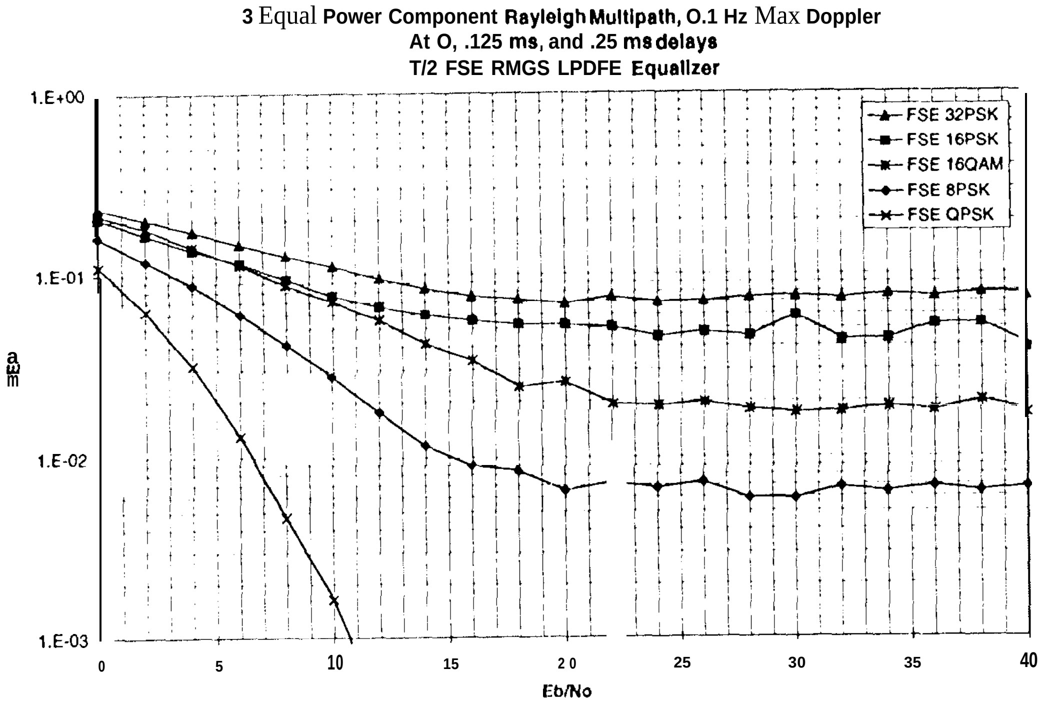


Figure 3. FSE Performance with Lower Max. Doppler Parameter (O. 1 Hz)

The Reed Solomon (255, 223) code is a good, low overhead, candidate code for this. As can bc seen from Figure 4, shows that the code starts correcting errors at a bit error rate of approximately 1 E-2, and at an error rate of 8E-3 reduces bit errors almost to 1E-3.

The (255,223) code corrects up to 16 byte errors pcr 255 byte block. Reed Solomon is a good candidate code for this application because it is possible to gradually increase the number of errors that can be corrected, at tbc expense of a slow increase in overhead. This is accomplished by devoting a larger portion of the block to check bytes. For example a (255,215) code would correct up to 20 byte errors.

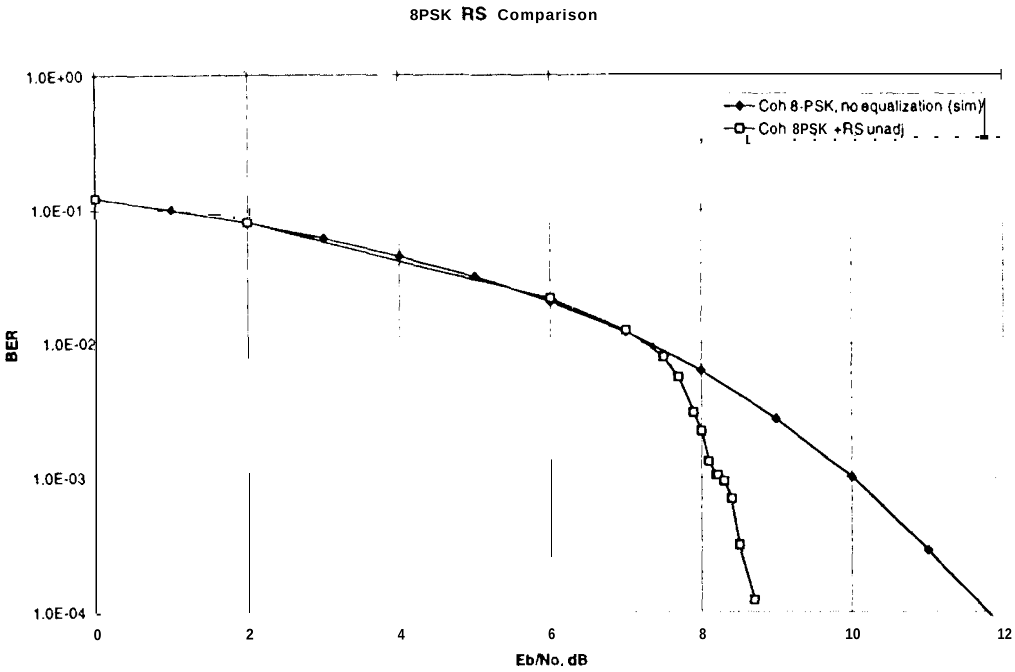


Figure 4. Measured Error Correction Performance Using Reed-Solomon (255, 223) Code

IV. PROPAGATION MEASUREMENTS AND PERFORMANCE TESTS

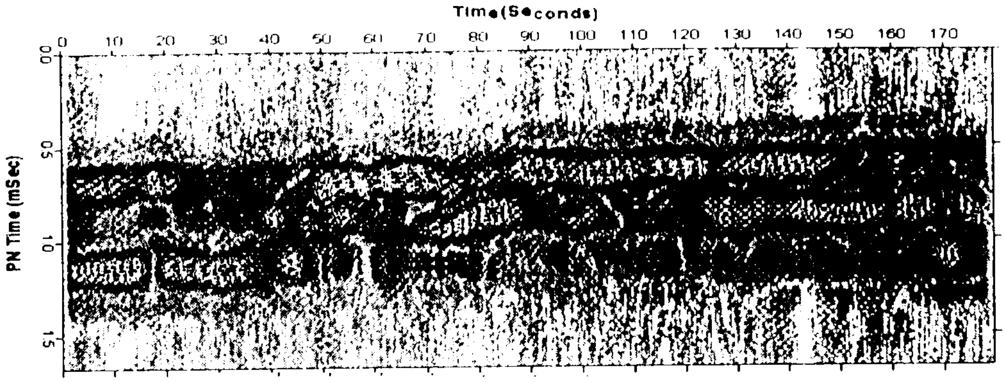
Propagation measurements and performance tests were performed using transmitters at the VOA Delano California facility. Test signal sequences were made up from a variety of modulation formats. These included BPSK pseudo noise (PN) sequences for measuring delay spread, compressed audio with various coding and modulation formats, and DSB AM to provide a qualitative assessment of the link.

The software modulator, running on a PC, was interfaced with a transmitter using digital to analog converters. Both linear 50 kW and non-linear 250 kW transmitters were used. To interface with the non-linear transmitter required that the signal be split into a constant amplitude, phase modulated carrier and an amplitude component. The phase modulated carrier was fed into the transmitter exciter, while the amplitude component was applied to the transmitter AM modulation input.

Initial propagation measurements took place in October 1996, with simultaneous reception taking place in Austin Texas, Washington DC, and Tenerife Spain. A second propagation and performance test was conducted in May 1997, to Washington DC, using a 10 kW linear transmitter. The third test took place in September 1997, also to Washington DC, using a non-linear transmitter at power levels of 220 kW and 50 kW. All of the above tests were. accomplished at two different frequencies to test different propagation modes between the transmitter and receive site.

The propagation measurements showed that at least some multipath always exists. In some cases there was a strong signal component with weaker distinct or diffuse multipath components, while in other cases there were several distinct, equal power components. The measured delay spread was in the range of 1 to 3 milliseconds.

The results of one aspect of the May 1997 test will be used to illustrate the propagation and performance data that was obtained. This was a transmission at 10 kW to Washington DC at 17.895 MHz. Figure 5 shows the results of the delay spread measurement, which shows three distinct, equal power signal components arriving approximately .25 milliseconds apart. This is a confirmation that something close to the classic three Rayleigh ray propagation model actually exists in nature.



Figure' 5. Delay Spread of Delano-Washington Path at 17.895 MHz in May 1997

Figure 6 shows the recorded spectrum at the receiver IF (centered at 13 kHz) during this 4 minute long test segment, Figure 7 shows the plot of the byte errors per code block (any number of errors in 8 bits) prior to the Reed Solomon decoder. This measurement is generated by the decoder. Since the (255, 223) code can correct up to 16 byte errors, anything between 1 and 16 byte errors in a 255 byte block will be corrected, If there are more than 16 byte errors in a block, the decoder gives an indication of -1 and does no error correction.

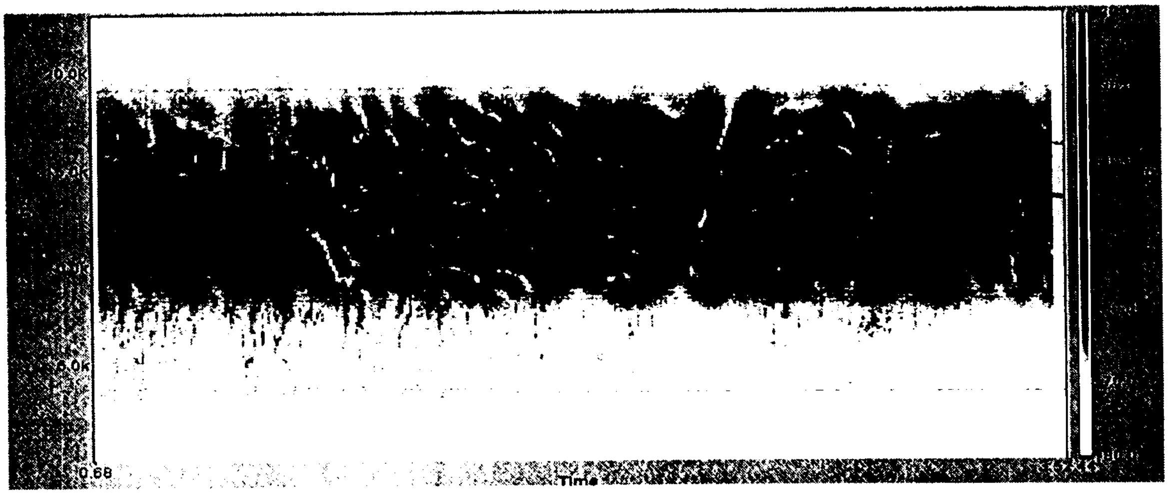


Figure 6. Received Spectrum (Centered at 13 kHz) of 8PSK Encoded Audio Segment

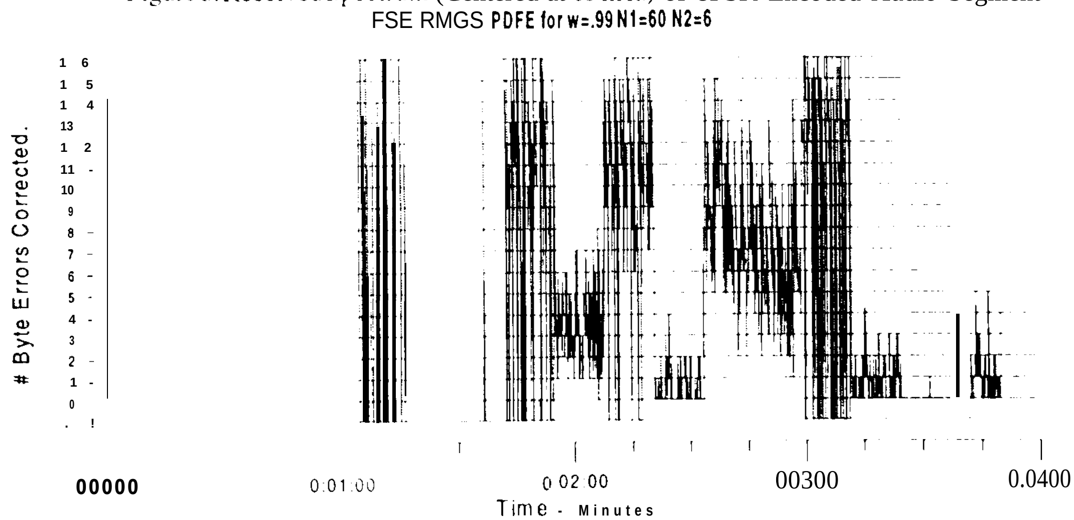


Figure 7. Decoder Performance, Delano to Washington, May 1997, 10kW, 8PSK, RS Code (255, 223)

It can be seen from Figure 7 that the decoder could not correct most blocks during the first half of this segment, but corrected most during the second half. A careful look at Figure 6 shows noise appearing outside of the 10 kHz signal spectrum during [he first half of the segment. This indicates that the signal power was low and the receiver AGC had brought the signal and noise up to the point the out of band noise is visible in the spectrum plot.

In this case the data being transmitted was compressed audio, so the recovered audio from the audio decoder was distorted during the first half and clear during the second half. Listening to the DSB AM segment that was transmitted as part of the test sequence indicates that the link was of very poor quality.

V. SUMMARY AND CONCLUSIONS

A system for digital audio transmission in the short wave bands was developed and tested over the air. Successful transmission of compressed audio with up to 8PSK modulation was accomplished during several tests. Link conditions during most of the tests were severe, as evaluated by listening to the DSB AM segments. Causes included low SNR, multipath, and interference from other transmissions.

Transmission at 8PSK in a 10 kHz channel allows a data rate of approximately 16 kbps, which is near AM quality. Further testing is needed under more benign conditions to determine if higher modulation levels and therefore higher data rates are practical. Having a hardware receiver, but still flexible in some of signal parameters, would be beneficial and provide faster turn-around between field trials and performance evaluation. In the mean time, some further conclusions in areas such as optimum coding overhead and effectiveness of time interleaving can be accomplished by additional processing of the recorded data.

IV. ACKNOWLEDGMENT

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V. REFERENCES

[1] Vaisnys, A,, and Chen, E., *Digital Audio Applications to Short Wave Broadcasting,* IEEE International Symposium on Industrial Electronics, Guimaraes, Portugal, 7-11 July 1997.